

AUTOMATIC SOUND FIELD CORRECTING SYSTEM AND  
A SOUND FIELD CORRECTING METHOD

BACKGROUND OF THE INVENTION

5        The present invention relates to an automatic sound field  
correcting system and a sound field correcting method for  
automatically correcting a sound field characteristic in an  
audio system having a plurality of loudspeakers.

10      The audio system that is equipped with a plurality of  
loudspeakers to provide a high quality sound field space is  
required to produce automatically the proper sound field space  
that can give a presence. In other words, when the listener  
tries to get the proper sound field space by himself or herself  
by operating the audio system, it is extremely hard to properly  
15      adjust a phase characteristic, a frequency characteristic, a  
sound pressure level, etc. of a reproduced sound that is played  
back via a plurality of loudspeakers. For this reason, it is  
required to correct automatically the sound field  
characteristic on the audio system side.

20      In the prior art, as the audio system of this type, the  
audio system disclosed in the Japanese Utility Model  
Application Publication No. Hei 6-13292 has been known. In  
this audio system in the prior art, an equalizer that receives  
audio signals on a plurality of channels to adjust these  
25      frequency characteristics of respective audio signals and a

plurality of delay circuits that delay the audio signals output from the equalizer every channel are provided, and then outputs of respective delay circuits are supplied to a plurality of loudspeakers.

5 Also, in order to correct the sound field characteristic,  
there are provided a pink noise generator, an impulse generator,  
a selector circuit, a microphone used to measure the reproduced  
sound being reproduced by the loudspeakers, a frequency  
analyzing means, and a delay time calculating means. Then,  
10 a pink noise generated by the pink noise generator is supplied  
to the equalizer via the selector circuit, and an impulse signal  
generated by the impulse generator is directly supplied to the  
loudspeakers via the selector circuit.

Upon correcting the phase characteristic of the sound field space, propagation delay times of the impulse sound from the loudspeakers to a listening position are measured by measuring the impulse sound reproduced via the loudspeakers by the microphone while supplying directly the impulse signal from the above impulse generator to the loudspeakers and then analyzing the measured signals by using the delay time calculating means.

In other words, the propagation delay times of respective impulse sounds are measured by directly supplying the impulse signal to individual loudspeakers while shifting a time and calculating time differences from points of time when

respective impulse signals are supplied to respective loudspeakers to points of time when respective impulse sounds being reproduced by every loudspeaker come up to the microphone by using the delay time calculating means. Thus, the phase  
5 characteristic of the sound field space can be corrected by adjusting the delay times of respective channels of the above delay circuit based on respective measured propagation delay times

Also, upon correcting the frequency characteristic of  
10 the sound field space, the pink noise is supplied from the pink noise generator to the equalizer and then the reproduced sounds of the pink noise being reproduced via a plurality of loudspeakers are measured by the microphone, and then frequency characteristics of these measured signals are analyzed by the  
15 frequency analyzing means. Thus, the frequency characteristic of the sound field space can be corrected by feedback-controlling the frequency characteristic of the equalizer based on the analyzed results.

20 SUMMARY OF THE INVENTION

In the audio system in the prior art, as described above, in order to correct the frequency characteristic of the sound field space, such a method is employed that the frequency characteristics of the reproduced sounds of the pink noise are  
25 analyzed by using a group of narrow-band filters and then the

analyzed results are fed back to the equalizer.

Now, upon producing the reproduced sounds of the pink noise, the pink noise is supplied to the equalizer after the frequency characteristic of the equalizer is set to a frequency 5 characteristic which mates with the audio playback. Accordingly, the reproduced sounds of the pink noise being reproduced via a plurality of loudspeakers reach the microphone and then the frequency characteristics of the reproduced sound 10 of the pink noise are analyzed by a group of narrow-band filters.

However, in case the frequency characteristics of measured signals derived from the reproduced sounds of the pink noise being reproduced via a plurality of (all) loudspeakers are frequency-analyzed by individual narrow- band filters in 15 a group of narrow-band filters, the analyzed result suitable for the frequency characteristic of the equalizer cannot be obtained with good precision. As a result, there is such a subject that, if the frequency characteristic of the equalizer is feedback-controlled based on the analyzed result, it becomes 20 difficult to correct properly the frequency characteristic of the sound field space.

In addition, there is such another subject that, since the phase characteristic of the sound field space is corrected based on the delay times that are obtained by supplying directly 25 the impulse signal to the loudspeakers, the phase

characteristic of the overall audio system cannot be corrected into the phase characteristic that can produce the proper sound field space.

It is an object of the present invention to overcome the above subjects in the prior art, and provide an automatic sound field correcting system and a sound field correcting method capable of providing a higher quality sound field space.

An automatic sound field correcting system of the present invention is an automatic sound field correcting system in an audio system for supplying a plurality of input audio signals to a plurality of sound generating means via a plurality of signal transmission lines, each of the plurality of signal transmission lines including an equalizer for adjusting a frequency characteristic of the audio signal, a channel-to-channel level adjusting means for adjusting a level of the audio signal, and a delaying means for adjusting a delay time of the audio signal, whereby the input audio signals are supplied to the sound generating means via the equalizers, the channel-to-channel level adjusting means, and the delaying means, the correcting system comprising a noise generating means for supplying a noise to respective signal transmission lines independently in correcting a sound field; detecting means for detecting reproduced sounds of the noise reproduced by the sound generating means; frequency characteristic correcting means for correcting frequency characteristics of

the equalizers based on detection results of the detecting means; channel-to-channel level correcting means for correcting an adjusted amount of the plurality of channel-to-channel level adjusting means based on the detection results 5 of the detecting means; and phase characteristic correcting means for calculating phase characteristics of the reproduced sounds reproduced by the sound generating means based on the detection results of the detecting means and also correcting delay times of the delaying means based on calculated phase 10 characteristics.

In the automatic sound field correcting system having such configuration, the equalizers, the channel-to-channel level adjusting means, and the delaying means are provided in the signal transmission lines via which the audio sound is 15 reproduced.

In such configuration, in correcting the sound field, the noise is supplied from the noise generating means to the equalizers every signal transmission line, and then respective reproduced sounds generated correspondingly are detected by 20 the detecting means. The frequency characteristic correcting means corrects the frequency characteristics of the equalizers based on detection results of the detecting means. Also, since the channel- to-channel level correcting means corrects the adjusted amount of the channel-to-channel level adjusting 25 means based on the detection results of the detecting means,

the levels of the audio signals supplied to respective sound generating means between the so-called channels are corrected precisely. In addition, since the phase characteristic correcting means corrects the delay times of the delaying means

5 based on the detection results of the detecting means, the phase characteristics of the audio signals supplied to respective sound generating means are corrected.

Accordingly, the frequency characteristic and the phase characteristic of the audio signals that are supplied to

10 respective sound generating means can be automatically and precisely corrected in reproducing the audio sound. Also, the rationalization of the phase characteristic and the frequency characteristic of the reproduced sounds reproduced by respective sound generating means at the listening position  
15 can be achieved. Therefore, the high quality sound field space with the presence can be provided.

In particular, in correcting the sound field, the noise is supplied to the sound generating means via the equalizers, the channel-to-channel level adjusting means, and the delaying

20 means, via which the audio sounds are reproduced, and then the equalizers, the channel-to-channel level adjusting means, and the delaying means are corrected based on measured results of the reproduced sounds of the noise reproduced by the sound generating means. Therefore, the correction of the sound  
25 field can be performed under the same condition as the

reproduction of the audio sound. For this reason, the sound field correction can be executed while totally taking account of the characteristic of the overall audio system and the characteristic of the sound field space.

5        Also, an automatic sound field correcting system of the present invention is an automatic sound field correcting system in an audio system for supplying a plurality of input audio signals to all frequency band sound generating means and a low frequency band exclusively reproducing sound generating means  
10      via a plurality of signal transmission lines, each of the plurality of signal transmission lines including an equalizer for adjusting a frequency characteristic of the audio signal, a channel-to-channel level adjusting means for adjusting a level of the audio signal, and a delaying means for adjusting  
15      a delay time of the audio signal, whereby the input audio signals are supplied to the sound generating means via the equalizers, the channel-to-channel level adjusting means, and the delaying means, the correcting system comprising a noise generating means for supplying a noise to the respective signal  
20      transmission lines independently in correcting a sound field; detecting means for detecting reproduced sounds of the noise reproduced by the sound generating means; frequency characteristic correcting means for correcting frequency characteristics of the equalizers based on detection results  
25      of the detecting means; first channel-to-channel level

correcting means for correcting an adjusted amount of the plurality of channel-to-channel level adjusting means of the signal transmission lines, in which the all frequency band sound generating means are provided, out of the plurality of

5 channel-to-channel level adjusting means based on the detection results of the detecting means; phase characteristic correcting means for calculating phase characteristics of the reproduced sounds reproduced by respective sound generating means based on the detection results of the detecting means

10 and also correcting delay times of the delaying means based on calculated phase characteristics; and second channel-to-channel level correcting means for correcting an adjusted amount of the plurality of channel-to-channel level adjusting means of the signal transmission lines, in which the low

15 frequency band exclusively reproducing sound generating means are provided, based on the detection results of the detecting means.

Also, the second channel-to-channel level correcting means corrects an adjusted amount of the channel-to-channel level adjusting means such that a sum of a spectrum average level of the reproduced sound reproduced by all frequency band sound generating means in the low frequency band and a spectrum average level of the reproduced sound reproduced by a low frequency band exclusively reproducing sound generating means

25 in the low frequency band and a spectrum average level of the

reproduced sound reproduced by the all frequency band sound generating means in the middle/high frequency band are made equal to a ratio of target curve data.

According to the automatic sound field correcting system  
5 having such configuration, since the correction of the sound field can be carried out under the same condition as the reproduction of the audio sound, the correction of the sound field can be implemented while totally taking account of the characteristic of the overall audio system and the  
10 characteristic of the sound field environment, and in addition the first channel-to-channel level correcting means corrects the adjusted amount of the channel-to-channel level adjusting means for the all frequency band sound generating means and also the second channel-to-channel level correcting means  
15 corrects the adjusted amount of the channel-to-channel level adjusting means for the low frequency band exclusively reproducing sound generating means, whereby the levels of the reproduced sounds reproduced by the all frequency band sound generating means and the low frequency band exclusively  
20 reproducing sound generating means can be made flat over the full audio frequency band.

Accordingly, generation of the sound field space to generate a feeling of physical disorder such that the levels of the reproduced sounds reproduced by the low frequency band  
25 exclusively reproducing sound generating means in the low

frequency band and the reproduced sounds reproduced by the all frequency band sound generating means in the all frequency band are enhanced or weakened at a certain frequency can be prevented, and also the high quality sound field space with the presence 5 can be implemented.

Also, a sound field correcting method of the present invention is a sound field correcting method in an audio system including a plurality of signal transmission lines for supplying a plurality of input audio signals separately to all 10 frequency band sound generating means and a low frequency band exclusively reproducing sound generating means, each of the plurality of signal transmission lines including an equalizer for adjusting a frequency characteristic of the audio signal, a channel-to-channel level adjusting means for adjusting a 15 level of the audio signal, and a delaying means for adjusting a delay time of the audio signal, whereby the input audio signals are supplied to the sound generating means via the equalizers, the channel-to-channel level adjusting means, and the delaying means, the method comprising a first step of 20 measuring reproduced sounds reproduced by the all frequency band sound generating means and a low frequency band exclusively reproducing sound generating means by inputting a noise, and then correcting frequency characteristics of the equalizers based on measured results; a second step of 25 measuring the reproduced sounds reproduced by the all frequency

band sound generating means and a low frequency band exclusively reproducing sound generating means by inputting the noise, and then correcting an adjusted amount of the channel-to-channel level adjusting means for the all frequency

5 band sound generating means based on the measured results; a third step of measuring the reproduced sounds reproduced by the all frequency band sound generating means and a low frequency band exclusively reproducing sound generating means by inputting the noise, and then correcting delay times of the

10 delaying means based on the measured results; a fourth step of measuring independently reproduced sounds reproduced by the all frequency band sound generating means and reproduced sounds reproduced by the low frequency band exclusively reproducing sound generating means; and a fifth step of correcting an

15 adjusted amount of the channel-to-channel level adjusting means based on measured results measured by the fourth step such that a sum of a spectrum average level of the reproduced sounds reproduced by the all frequency band sound generating means in a low frequency band and a spectrum average level of

20 the reproduced sound reproduced by a low frequency band exclusively reproducing sound generating means in the low frequency band and a spectrum average level of the reproduced sound reproduced by the all frequency band sound generating means in a middle/high frequency band are set equal to a ratio

25 of target curve data. Accordingly, generation of the sound

field space to generate a feeling of physical disorder such that the levels of the reproduced sounds reproduced by the low frequency band exclusively reproducing sound generating means in the low frequency band and the reproduced sounds reproduced

5 by the all frequency band sound generating means in the all frequency band are enhanced or weakened at a certain frequency can be prevented, and also the high quality sound field space with the presence can be accomplished.

According to the sound field correcting method of the  
10 present invention, since the correction of the sound field can be carried out under the same condition as the reproduction of the audio sound, the correction of the sound field can be implemented while totally taking account of the characteristic of the overall audio system and the characteristic of the sound  
15 field environment, and in addition the levels of the reproduced sounds reproduced by the all frequency band sound generating means and the low frequency band exclusively reproducing sound generating means can be made flat over the full audio frequency band. Accordingly, generation of the sound field space to  
20 generate the feeling of physical disorder such that the levels of the reproduced sounds reproduced by the low frequency band exclusively reproducing sound generating means in the low frequency band and the reproduced sounds reproduced by the all frequency band sound generating means in the all frequency band  
25 are enhanced or weakened at a certain frequency can be prevented,

and thus the high quality sound field space with the presence can be implemented.

BRIEF DESCRIPTION OF THE DRAWINGS

5 FIG.1 is a block diagram showing a configuration of an audio system including an automatic sound field correcting system according to the present embodiment.

10 FIG.2 is a block diagram showing a configuration of the automatic sound field correcting system according to the present embodiment.

FIG.3 is a block diagram showing a pertinent configuration of the automatic sound field correcting system according to the present embodiment.

15 FIG.4 is a block diagram showing another pertinent configuration of the automatic sound field correcting system according to the present embodiment.

FIG.5 is a view showing a frequency characteristic of a graphic equalizer.

20 FIG.6 is a view showing the problem in a low frequency band of a reproduced sound.

FIG.7 is a view showing an example of arrangement of loudspeakers.

25 FIG.8 is a flowchart showing an operation of the automatic sound field correcting system according to the present embodiment.

FIG.9 is a flowchart showing a frequency characteristic correcting process.

FIG.10 is a flowchart showing a channel-to-channel level correcting process.

5 FIG.11 is a flowchart showing a phase characteristic correcting process.

FIG.12 is a flowchart showing a flatness correcting process.

10 DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

An embodiment of an automatic sound field correcting system of the present invention will be explained with reference to the accompanying drawings hereinafter. FIG.1 is a block diagram showing a configuration of an audio system including the automatic sound field correcting system according to the present embodiment. FIG.2 to FIG.4 are block diagrams showing the configuration of the automatic sound field correcting system.

In FIG.1, a signal processing circuit 2 to which digital audio signals  $S_{FL}$ ,  $S_{FR}$ ,  $S_C$ ,  $S_{RL}$ ,  $S_{RR}$ ,  $S_{WF}$  are supplied from a sound source 1 such as a CD (Compact Disk) player, a DVD (Digital Video Disk or Digital Versatile Disk) player, etc. via a signal transmission line having a plurality of channels, and a noise generator 3 are provided to the present audio system.

25 Also, D/A converters  $4_{FL}$ ,  $4_{FR}$ ,  $4_C$ ,  $4_{RL}$ ,  $4_{RR}$ ,  $4_{WF}$  for converting

digital outputs  $D_{FL}$ ,  $D_{FR}$ ,  $D_C$ ,  $D_{RL}$ ,  $D_{WF}$  which are signal-processed by the signal processing circuit 2 into analog signals, and amplifiers  $S_{FL}$ ,  $S_{FR}$ ,  $S_C$ ,  $S_{RL}$ ,  $S_{RR}$ ,  $S_{WF}$  for amplifying respective analog audio signals being output from these D/A converters 5 are provided. Respective analog audio signals  $SP_{FL}$ ,  $SP_{FR}$ ,  $SP_C$ ,  $SP_{RL}$ ,  $SP_{RR}$ ,  $SP_{WF}$  amplified by these amplifiers are supplied to loudspeakers  $5_{FL}$ ,  $5_{FR}$ ,  $5_C$ ,  $5_{RL}$ ,  $5_{RR}$ ,  $5_{WF}$  on a plurality of channels arranged in a listening room 7, etc., as shown in FIG.7, to sound them.

10 In addition, a microphone 8 for collecting reproduced sounds at a listening position RV, an amplifier 9 for amplifying a sound collecting signal SM output from the microphone 8, and an A/D converter 10 for converting an output of the amplifier 9 into digital sound collecting data DM to supply to the signal processing circuit 2 are provided.

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Then, the present audio system provides a sound field space with a presence to the listener at the listening position RV by sounding all frequency band type loudspeakers  $6_{FL}$ ,  $6_{FR}$ ,  $6_C$ ,  $6_{RL}$ ,  $6_{RR}$  each has a frequency characteristic that enables 20 an almost full range of the audio frequency band to reproduce, and a low frequency band exclusively reproducing loudspeaker  $6_{WF}$  that has a frequency characteristic to reproduce only the so-called heavy and low sound.

For example, as shown in FIG.7, in the case that the 25 listener arranges the front loudspeakers (front left-side

loudspeaker, front right-side loudspeaker)  $6_{FL}$ ,  $6_{FR}$  on two right and left channels and the center loudspeaker  $6_C$  in front of the listening position RV, arranged the rear loudspeakers (rear left-side loudspeaker, rear right-side loudspeaker)  $6_{RL}$ ,  $6_{RR}$  on 5 two right and left channels at the rear of the listening position RV, and arranges the low frequency band exclusively reproducing subwoofer  $6_{WF}$  at any position according to his or her taste, the automatic sound field correcting system installed in the present audio system can implement the sound 10 field space with the presence by sounding six loudspeakers  $6_{FL}$ ,  $6_{FR}$ ,  $6_C$ ,  $6_{RL}$ ,  $6_{RR}$ ,  $6_{WF}$  by supplying the analog audio signals  $SP_{FL}$ ,  $SP_{FR}$ ,  $SP_C$ ,  $SP_{RL}$ ,  $SP_{RR}$ ,  $SP_{WF}$ , whose frequency characteristic and phase characteristic are corrected, to these loudspeakers.

In this case, in the following explanation, respective 15 channels are denoted by numbers  $x$  ( $1 \leq x \leq k$ ).

The signal processing circuit 2 is composed of a digital signal processor (DSP), or the like, and comprises a graphic equalizer GEQ and channel-to-channel attenuators ATG<sub>1</sub> to ATG<sub>k</sub>, and delay circuits DLY<sub>1</sub> to DLY<sub>k</sub>, that are shown in FIG.2, and 20 a frequency characteristic correcting portion 11, a channel-to-channel level correcting portion 12, a phase characteristic correcting portion 13 and a flatness correcting portion 14, that are shown in FIG.3.

The frequency characteristic correcting portion 11 25 adjusts frequency characteristics of equalizers EQ<sub>1</sub> to EQ<sub>k</sub> on

respective channels of the graphic equalizer. The channel-to-channel level correcting portion 12 and the flatness correcting portion 14 adjust the attenuation factors of the channel-to-channel attenuators ATG<sub>1</sub> to ATG<sub>k</sub>. The phase 5 characteristic correcting portion 13 adjusts delay times of the delay circuits DLY<sub>1</sub> to DLY<sub>k</sub>, whereby the sound field correction is performed.

As shown in a frequency characteristic diagram of FIG.5, the equalizers EQ<sub>1</sub> to EQ<sub>5</sub> on first to fifth channels (x=1 to 10 5) are constructed such that their frequency characteristics can be precisely adjusted for respective j frequencies f<sub>1</sub> to f<sub>j</sub>. More particularly, respective frequencies f<sub>1</sub> to f<sub>i</sub> in FIG.5 are decided by dividing the low frequency band below about 0.2 kHz into about five ranges, and then respective frequencies 15 f<sub>i+1</sub> to f<sub>j</sub> are decided by dividing the middle/high frequency band in excess of about 0.2 kHz into about thirteen ranges. Then, the frequency characteristics can be precisely adjusted by adjusting filter coefficients of the equalizers EQ<sub>1</sub> to EQ<sub>5</sub>, based on filter coefficient adjust signals SF<sub>1</sub> to SF<sub>5</sub>.

20 The equalizer EQ<sub>k</sub> on the k-th channel is constructed to adjust the frequency characteristic in the low frequency band. Then, the frequency characteristic below about 0.2 kHz shown in FIG.5 can be precisely adjusted for respective frequencies f<sub>1</sub> to f<sub>i</sub> by adjusting the filter coefficient of the equalizer 25 EQ<sub>k</sub> based on filter coefficient adjust signal SF<sub>k</sub>.

Also, a switch element  $SW_{12}$  that ON/OFF-controls an input of the digital audio signal  $S_{FL}$  from the sound source 1 and a switch element  $SW_{11}$  that ON/OFF-controls an input of a noise signal  $DN$  from the noise generator 3 are connected the equalizer

5  $EQ_1$  on the first channel. Also, the switch element  $SW_{11}$  is connected to the noise generator 3 via a switch element  $SW_N$ .

The switch elements  $SW_{11}$ ,  $SW_{12}$ ,  $SW_N$  are controlled by a system controller MPU that consists of a microprocessor shown in FIG.3. At the time of reproducing the audio sound, the 10 switch element  $SW_{12}$  is turned ON (conductive) and the switch elements  $SW_{11}$ ,  $SW_N$  are turned OFF (nonconductive). At the time of correcting the sound field, the switch element  $SW_{12}$  is turned OFF and the switch elements  $SW_{11}$ ,  $SW_N$  are turned ON.

In addition, the channel-to-channel attenuator  $ATG_1$  is 15 connected to an output contact of the equalizer  $EQ_1$ , and also the delay circuit  $DLY_1$  is connected to an output contact of the channel-to-channel attenuator  $ATG_1$ . Then, an output  $D_{FL}$  of the delay circuit  $DLY_1$  is supplied to the D/A converter  $4_{FL}$  in FIG.1.

20 The second to  $k$ -th channels have similar configuration to the first channel, and include switch elements  $SW_{21}$  to  $SW_{k1}$  corresponding to the switch element  $SW_{11}$  and switch elements  $SW_{22}$  to  $SW_{k2}$  corresponding to the switch element  $SW_{12}$  respectively. Then, the equalizers  $EQ_1$  to  $EQ_k$ , the channel- to-channel 25 attenuators  $ATG_2$  to  $ATG_k$ , and the delay circuits  $DLY_1$  to  $DLY_k$

are provided following to these switch elements  $SW_{22}$  to  $SW_{k2}$ .

Then, output  $D_{FR}$  to  $D_{WF}$  of the delay circuits  $DLY_2$  to  $DLY_k$  are supplied to the D/A converters  $4_{FR}$  to  $4_{WF}$  in FIG.1.

Further, the channel-to-channel attenuators  $ATG_1$  to  $ATG_5$

5 on the first to fifth channels change their attenuation factors in the range of 0 dB to the (-) side in compliance with the adjust signals  $SG_1$  to  $SG_5$  from the channel-to- channel level correcting portion 12 respectively. Also, the channel-to-channel attenuator  $ATG_k$  on the k-th channel changes its  
10 attenuation factor in the range of 0 dB to the (-) side in compliance with the adjust signal  $SG_k$  from the flatness correcting portion 14.

The delay circuit  $DLY_1$  to  $DLY_k$  on the first to k-th channels change their delay times in compliance with the adjust

15 signal  $SDL_1$  to  $SDL_k$  from the phase characteristic correcting portion 13.

As shown in FIG.4, the frequency characteristic correcting portion 11 is constructed to have a band-pass filter

11a, a coefficient table 11b, a gain calculating portion 11c,  
20 a coefficient deciding portion 11d, and a coefficient table 11e.

The band-pass filter 11a is composed of narrow-band digital filters that have the frequencies  $f_1$  to  $f_j$  set by the equalizers  $EQ_1$  to  $EQ_k$  as their center frequencies respectively,  
25 and supplies data  $[PxJ]$  indicating levels of respective

frequencies  $f_1$  to  $f_j$  to the gain calculating portion 11c by frequency-discriminating the sound collecting data DM from the D/A converter 10 for respective frequencies  $f_1$  to  $f_j$ . In this case, the frequency discriminating characteristic of the band-pass filter 11a is set by the filter coefficient data stored previously in the coefficient table 11b.

The gain calculating portion 11c calculates gains of the equalizers  $EQ_1$  to  $EQ_k$  for respective frequencies  $f_1$  to  $f_k$  in correcting the sound field based on the data  $[PxJ]$  indicating the above levels, and then supplies calculated gain data  $[GxJ]$  to the coefficient deciding portion 11d. In other words, the gains of the equalizers  $EQ_1$  to  $EQ_k$  are counted back for respective frequencies  $f_1$  to  $f_k$  by using the data  $[PxJ]$  as the already-known transfer functions of the equalizers  $EQ_1$  to  $EQ_k$ .

The coefficient deciding portion 11d generates the filter coefficient adjust signals  $SF_1$  to  $SF_s$  to adjust the frequency characteristics of the equalizers  $EQ_1$  to  $EQ_s$  under control of the system controller MPU. Upon correcting the sound field, the coefficient deciding portion 11d generates the filter coefficient adjust signals  $SF_1$  to  $SF_s$  according to the conditions instructed by the listener.

If the standard sound field correction being set previously in the present sound field correcting system is performed without the instruction for the conditions of the correction of the sound field by the listener, the filter

coefficient data to adjust the frequency characteristics of the equalizers  $EQ_1$  to  $EQ_k$  are read from the coefficient table 11e based on the gain data  $[GxJ]$  supplied from the gain calculating portion 11c for respective frequencies  $f_1$  to  $f_k$ ,  
5 and then the frequency characteristics of the equalizers  $EQ_1$  to  $EQ_k$  are adjusted by the filter coefficient adjust signals  $SF_1$  to  $SF_k$  of the filter coefficient data.

In other words, the filter coefficient data to adjust variously the frequency characteristics of the equalizers  $EQ_1$  to  $EQ_k$  are stored previously as look-up tables in the coefficient table 11e. Then, the coefficient deciding portion 11d reads the filter coefficient data corresponding to the gain data  $[GxJ]$  and then supplies such read filter coefficient data as the filter coefficient adjust signals  $SF_1$  to  $SF_k$  to the equalizers  $EQ_1$  to  $EQ_k$ , to thereby adjust the frequency characteristic every channel.  
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If the listener selects the target curve, described later, to perform the correction of the sound field, the coefficient deciding portion 11d memory-accesses the target curve data  $TGx$  stored previously in the coefficient table 11e and also memory-accesses the filter coefficient data corresponding to the gain data  $[GxJ]$  supplied from the gain calculating portion 11c. Then, the coefficient deciding portion 11d generates the filter coefficient data, that are modulated by the target curve data  $TGx$ , by executing predetermined calculations based on the  
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target curve data  $TG_x$  and the filter coefficient data, and then supplies the filter coefficient data as the filter coefficient adjust signals  $SF_1$  to  $SF_5$  to the equalizers  $EQ_1$  to  $EQ_5$ , to thereby adjust the frequency characteristic every channel.

5 In this case, the target curve denotes the frequency characteristic of the reproduced sound that can suit the listener's taste. In the present audio system, in addition to the target curve used to generate the reproduced sound having the frequency characteristic that is suitable for the classic 10 music, various target curve data used to generate the reproduced sounds having the frequency characteristics that are suitable for rock music, pops, vocal, etc. are stored.

15 The channel-to-channel level correcting portion 12 receives respective sound collecting data  $DM$  obtained when all frequency band loudspeakers  $6_{FL}$ ,  $6_{FR}$ ,  $6_C$ ,  $6_{RL}$ ,  $6_{RR}$  are sounded individually by the noise signal (pink noise)  $DN$  output from the noise generator 3, and then measures the levels of the reproduced sounds of respective loudspeakers at the listening 20 position  $RV$  based on the sound collecting data  $DM$ . Then, the channel-to-channel level correcting portion 12 generates the 25 adjust signals  $SG_1$  to  $SG_5$  based on these measured results and corrects automatically the attenuation factors of the channel-to-channel attenuators  $ATG_1$  to  $ATG_5$  by the adjust signals  $SG_1$  to  $SG_5$ . The level adjustment (gain adjustment) between the first to fifth channels is carried out based on

the adjustment of the attenuation factors by the channel-to-channel level correcting portion 12.

However, the channel-to-channel level correcting portion 12 does not adjust the attenuation factor of the 5 channel-to-channel attenuator  $ATG_k$ , but the flatness correcting portion 14 adjusts the attenuation factor of the channel-to-channel attenuator  $ATG_k$ .

The phase characteristic correcting portion 13 measures the phase characteristics of respective channels based on 10 respective sound collecting data  $DM$  obtained when respective loudspeakers  $6_{FL}$ ,  $6_{FR}$ ,  $6_c$ ,  $6_{RL}$ ,  $6_{RR}$ ,  $6_{WF}$  are sounded individually by the noise signal (pink noise)  $DN$  output from the noise generator 3, and then corrects the phase characteristic of the sound field space in compliance with the measured result.

15 More particularly, the loudspeakers  $6_{FL}$ ,  $6_{FR}$ ,  $6_c$ ,  $6_{RL}$ ,  $6_{RR}$ ,  $6_{WF}$  on respective channels are sounded by the noise signal  $DN$  every period  $T$ , and then cross correlations between resultant sound collecting data  $DM_1$ ,  $DM_2$ ,  $DM_3$ ,  $DM_4$ ,  $DM_5$ ,  $DM_k$  on respective channels are calculated. Here, the cross correlation between 20 the sound collecting data  $DM_2$  and  $DM_1$ , the cross correlation between the sound collecting data  $DM_3$  and  $DM_1$ , ..., the cross correlation between the sound collecting data  $DM_k$  and  $DM_1$  are calculated, and then peak intervals (phase differences) between respective correlation values are set as their delay 25 times  $\tau_2$  to  $\tau_k$  in respective system circuits  $CQT_2$  to  $CQT_k$ . That

is, the delay times  $\tau_2$  to  $\tau_k$  of remaining system circuits  $CQT_2$  to  $CQT_k$  are calculated on the basis of the phase of the sound collecting data  $DM1$  obtained from the system circuit  $CQT_1$  (i.e., phase difference 0,  $\tau_1=0$ ). Then, the adjust signals  $SDL_1$  to  $SDL_k$  are generated based on measured results of these delay times  $\tau_2$  to  $\tau_k$ , and then the phase characteristic of the sound field space is corrected by automatically adjusting respective delay times of the delay circuits  $DLY_1$  to  $DLY_k$  by using these adjust signals  $SDL_1$  to  $SDL_k$ . In this case, the pink noise is employed to correct the phase characteristic in the present embodiment, but the present invention is not limited to this noise and other noises may be employed.

The flatness correcting portion 14 adjusts the attenuation factor of the channel-to-channel attenuator  $ATG_k$ , that is not adjusted by the channel-to-channel level correcting portion 12, after the adjustments made by the frequency characteristic correcting portion 11, the channel-to-channel level correcting portion 12, and the phase characteristic correcting portion 13 have been completed.

Although their details are described later, the spectra of the reproduced sounds of the noise reproduced by the all frequency band loudspeakers  $6_{FL}$ ,  $6_{FR}$ ,  $6_c$ ,  $6_{RL}$ ,  $6_{RR}$  are detected by spectrum-analyzing the sound collecting data  $DM$  obtained when the all frequency band loudspeakers  $6_{FL}$ ,  $6_{FR}$ ,  $6_c$ ,  $6_{RL}$ ,  $6_{RR}$  except the loudspeaker  $6_{RF}$  are sounded simultaneously based on

the noise signal (uncorrelated noise) DN output from the noise generator 3, and in addition the spectrum of the reproduced sounds of the noise reproduced by the loudspeaker 6<sub>WF</sub> is detected by spectrum-analyzing the sound collecting data DM obtained 5 when only the low frequency band exclusively reproducing loudspeaker 6<sub>WF</sub> is sounded based on the noise signal (pink noise) DN output from the noise generator 3.

By executing predetermined calculations based on these spectra, there is generated the adjust signal SG<sub>k</sub> that makes 10 the frequency characteristic of the reproduced sound flat over all audio frequency bands when all loudspeakers 6<sub>FL</sub>, 6<sub>FR</sub>, 6<sub>C</sub>, 6<sub>RL</sub>, 6<sub>RR</sub>, 6<sub>WF</sub> are sounded simultaneously.

That is, as shown in the frequency characteristic diagram of FIG.6, since the all frequency band loudspeakers 6<sub>FL</sub>, 6<sub>FR</sub>, 15 6<sub>C</sub>, 6<sub>RL</sub>, 6<sub>RR</sub> have not only the middle/high frequency band reproducing capability but also the low frequency band reproducing capability, in some cases the levels of the low frequency sounds reproduced by the loudspeakers 6<sub>FL</sub>, 6<sub>FR</sub>, 6<sub>C</sub>, 6<sub>RL</sub>, 6<sub>RR</sub> and the low frequency sound reproduced by the 20 loudspeaker 6<sub>WF</sub>, for example, become higher than the level of the reproduced sound in the middle/high frequency band if these loudspeakers 6<sub>FL</sub>, 6<sub>FR</sub>, 6<sub>C</sub>, 6<sub>RL</sub>, 6<sub>RR</sub> and the low frequency band exclusively reproducing loudspeaker 6<sub>WF</sub> are sounded. Thus, there is caused such a problem that such low frequency sounds 25 are offensive to the ear and also give the listener an

unpleasant feeling. Therefore, the calculating portion 15d  
adjusts the attenuation factor of the channel- to-channel  
attenuator ATG<sub>k</sub> by the adjust signal SG<sub>k</sub> such that a sum of the  
spectrum average levels of the above low frequency band sounds  
5 and the spectrum average levels of the middle/high frequency  
band sounds are set equal to a ratio of the target  
characteristics (ratio of the target curve data).

In this case, the configuration of the automatic sound  
field correcting system is explained as above, but more  
10 detailed functions will be explained in detail in the  
explanation of operation.

Next, an operation of the automatic sound field  
correcting system having such configuration will be explained  
with reference to flowcharts shown in FIG.8 to FIG.12  
15 hereunder.

When the listener arranges a plurality of loudspeakers  
6<sub>RL</sub> to 6<sub>RF</sub> in the listening room 7, etc. and connects them to  
the present audio system, as shown in FIG.7, for example, and  
then instructs to start the sound field correction by operating  
20 a remote controller (not shown) provided to the present audio  
system, the system controller MPU operates the automatic sound  
field correcting system in compliance with this instruction.

First, an outline of the operation of the automatic sound  
field correcting system will be explained with reference to  
25 FIG.8. In the frequency characteristic correcting process in

step S10, the process for adjusting the frequency characteristics of the equalizers  $EQ_1$  to  $EQ_k$  is carried out by the frequency characteristic correcting portion 11.

5 In the channel-to-channel level correcting process in step S20, the process for adjusting the attenuation factors of the channel-to-channel attenuators  $ATG_1$  to  $ATG_5$  is carried out by the channel-to-channel level correcting portion 12. That is, in step S20, the channel- to-channel attenuator  $ATG_k$  on the k-th channel is not adjusted.

10 In the phase characteristic correcting process in step S30, the process for adjusting the delay times of the delay circuits  $DLY_1$  to  $DLY_k$  for all channels is carried out by the phase characteristic correcting portion 13.

15 In the flatness correcting process in step S40, the process for making the frequency characteristic of the reproduced sound at the listening position RV flat over the full audio frequency band is carried out by adjusting the attenuation factor of the channel-to-channel attenuator  $ATG_k$  on the k-th channel by using the flatness correcting portion 20 14.

In this manner, the present automatic sound field correcting system executes the sound field correction by performing in sequence the correcting processes that are classified roughly into four stages.

25 Then, operations in respective process stages will be

explained in detail in sequence.

First, the frequency characteristic correcting process in step S10 will be explained in detail. The process in step S10 will be carried out in compliance with the detailed flowchart shown in FIG.9.

In step S100, the initialization process is executed to make the frequency characteristics of the equalizers  $EQ_1$  to  $EQ_k$  flat by the filter coefficient adjust signals  $SF_1$  to  $SF_k$ .

That is, the gains of the equalizers  $EQ_1$  to  $EQ_k$  are set to 0 dB over the full audio frequency band. Also, the attenuation factors of the channel-to-channel attenuators  $ATG_1$  to  $ATG_k$  are set to 0 dB, and the delay times of all delay circuits  $DLY_1$  to  $DLY_k$  are set to 0, and the amplification factors of the amplifiers  $S_{fl}$  to  $S_{fr}$  shown in FIG.1 are set equal.

In addition, the switch elements  $SW_{12}$ ,  $SW_{22}$ ,  $SW_{32}$ ,  $SW_{42}$ ,  $SW_{52}$ ,  $SW_{k2}$  are turned OFF (nonconductive) to cut off the input from the sound source 1, and the switch element  $SW_N$  is turned ON (conductive). Accordingly, the signal processing circuit 2 is set to the state that the noise signal (pink noise)  $DN$  generated by the noise generator 3 is supplied to the equalizers  $EQ_1$  to  $EQ_k$ .

Then, in step S102, in case the listener selects the desired target curve, the frequency characteristics of the equalizers  $EQ_1$  to  $EQ_k$  are set based on the target curve data [TGx]. In contrast, in case the listener does not select the

target curve, the frequency characteristics of the equalizers EQ<sub>1</sub> to EQ<sub>k</sub> are set, as they are, to those in above initialization process.

As indicated by a matrix in following Eq.(1), the target curve data [TG<sub>x</sub>] consist of a plurality of data TG<sub>1</sub> to TG<sub>k</sub> for respective channels x (=1 to k), and are provided in plural to respond to types of the music such as the classic, the rock music, etc. The listener can select the target curve every channel or select in answer to the type of the music such as the classic, the rock music, etc. by operating a remote controller.

$$[T G x] = \begin{bmatrix} T G 1 \\ T G 2 \\ T G 3 \\ T G 4 \\ T G 5 \\ \vdots \\ T G k \end{bmatrix} \quad \dots (1)$$

Then, the process goes to step S104, and flag data n=0 is set in a flag register (not shown) built in the system controller MPU.

Then, the sound field characteristic measuring process is executed in step S106.

In this step, the noise signal DN is supplied in sequence to the first to k-th channels by exclusively turning ON the switch elements SW<sub>11</sub>, SW<sub>21</sub>, SW<sub>31</sub>, SW<sub>41</sub>, SW<sub>51</sub>, SW<sub>k1</sub> for the

predetermined period T respectively

Accordingly, the microphone 8 collects the noise sound that is produced in sequence by respective loudspeakers 6FL to 6WF on the first to k-th channels. Then, the sound 5 collecting data DM are supplied to the frequency characteristic correcting portion 11.

In addition, the sound collecting data DM for respective channels are frequency-divided by the band-pass filter 11a and then supplied to the gain calculating portion 11c. Therefore, 10 data [PxJ] represented by a matrix in following Eq.(2) are supplied to the gain calculating portion 11c.

$$15 \quad [P \times J] = \begin{bmatrix} P_{11} & \dots & P_{1j} \\ P_{21} & \dots & P_{2j} \\ P_{31} & \dots & P_{3j} \\ P_{41} & \dots & P_{4j} \\ P_{51} & \dots & P_{5j} \\ \vdots & \ddots & \vdots \\ P_{k1} & \dots & P_{kj} \end{bmatrix} \dots (2)$$

Then, in step S108, the gain calculating portion 11c spectrum-analyzes the data [PxJ] for respective channels. 20 Then, in step S110, the gains of the equalizers EQ<sub>1</sub> to EQ<sub>k</sub> are calculated based on these spectrum-analyzed results. Accordingly, gain data [G0xJ] represented by a matrix in following Eq.(3) are calculated and then supplied to the coefficient deciding portion 11d.

addition, the loudspeakers  $6_{FR}$ ,  $6_C$ ,  $6_{RL}$ ,  $6_{RR}$ ,  $6_{WF}$  on remaining channels are similarly decided.

Then, in step S112, it is decided whether or not the flag data  $n$  is 1. If NO, the flag data  $n$  is set to 1 in step S114, 5 and then the process goes to step S116.

In step S116, the coefficient deciding portion 11d acquires the filter coefficient data from the coefficient table 11e based on the gain data  $[G0xJ]$ , and then adjusts the frequency characteristic of the equalizers  $EQ_1$  to  $EQ_k$  by the 10 filter coefficient adjust signals  $SF1$  to  $SF_k$ .

Also, in above step S108, in case the channel in which the small loudspeaker is connected is decided, the frequency characteristic of the equalizer on the channel is adjusted to 0 dB. The frequency characteristic of the equalizer on the 15 channel in which the large loudspeaker is connected is adjusted based on the filter coefficient data obtained according to the above gain data  $[G0xJ]$ .

In the present embodiment, the size of the loudspeaker is decided by comparing the gain data  $[G0xJ]$  with the threshold 20 value  $THDCH$ . But the size of the loudspeaker may be decided by comparing the data  $[PxJ]$  obtained in the sound field characteristic measuring process in step S106 with the threshold value.

Then, after the process in step S116, the processes 25 starting from step S106 are repeated.

In this manner, the processes in step S106 and subsequent steps are repeated. In step S112, if it is decided that the flag data n is 1, the process goes to step S118.

If the processes in step S104 and subsequent steps are 5 repeated, the flag data n is set to n=1 and thus the calculations in above Eqs. (2) (3) are executed once again. Thus, the gain data  $[G1 \times J]$  represented by a matrix in following Eq.(4) corresponding to above Eq.(3) are calculated. In this case, 10 a suffix 1 of the gain data  $[G1 \times J]$  denotes the flag data n (=1),  $x$  denotes the number of the channel, and  $J$  denotes the order 1...i...j of the frequencies being set in the equalizers  $EQ_1$  to  $EQ_k$ .

15 
$$[G1 \times J] = \begin{bmatrix} G1(1, 1) & \dots & G1(1, j) \\ G1(2, 1) & \dots & G1(2, j) \\ G1(3, 1) & \dots & G1(3, j) \\ G1(4, 1) & \dots & G1(4, j) \\ G1(5, 1) & \dots & G1(5, j) \\ G1(k, 1) & \dots & G1(k, j) \end{bmatrix} \quad \dots (4)$$

Then, in step S118, the gain calculating portion 11c adds 20 the gain data  $[G0 \times J]$  and  $[G1 \times J]$  on above Eqs.(3)(4) for respective rows and columns to calculate the optimum gain data  $[GxJ]_{opt}$  represented by a matrix in following Eq.(5), and supplies them to the coefficient deciding portion 11d.

$$[G \times J]_{opt} = \begin{bmatrix} G_0(1, 1) + G_1(1, 1) & \dots & G_0(1, j) + G_1(1, j) \\ G_0(2, 1) + G_1(2, 1) & \dots & G_0(2, j) + G_1(2, j) \\ G_0(3, 1) + G_1(3, 1) & \dots & G_0(3, j) + G_1(3, j) \\ G_0(4, 1) + G_1(4, 1) & \dots & G_0(4, j) + G_1(4, j) \\ G_0(5, 1) + G_1(5, 1) & \dots & G_0(5, j) + G_1(5, j) \\ G_0(k, 1) + G_1(k, 1) & \dots & G_0(k, j) + G_1(k, j) \end{bmatrix} \quad \dots (5)$$

5        In addition, the coefficient deciding portion 11d  
 acquires the filter coefficient data from the coefficient table  
 11e based on the gain data  $[G \times J]_{opt}$ . Then, in step S120, the  
 frequency characteristics of the equalizers  $EQ_1$  to  $EQ_k$  are  
 10      finally adjusted by the filter coefficient adjust signals  $SF_1$   
 to  $SF_k$  based on the filter coefficient data.

In this way, the frequency characteristic of the sound  
 field space is corrected by adjusting the frequency  
 characteristics of the equalizers  $EQ_1$  to  $EQ_k$  by virtue of the  
 frequency characteristic correcting portion 11.

15      Also, in the sound field characteristic measuring  
 process in step S106, since respective loudspeakers  $6_{FL}$ ,  $6_{TR}$ ,  
 $6_C$ ,  $6_{RL}$ ,  $6_{RR}$ ,  $6_{WF}$  are sounded by the frequency-divided pink noise  
 and then resultant reproduced sounds are collected, the  
 frequency characteristics and the reproducing capabilities  
 20      (output powers) of respective loudspeakers can be detected.  
 Therefore, the total rationalization of the frequency  
 characteristic can be achieved while taking account of the  
 frequency characteristics and the reproducing capabilities of  
 respective loudspeakers.

25      Next, the channel-to-channel level correcting process

$$[G_{0 \times J}] = \left[ \begin{array}{c} G_0(1, 1) \dots G_0(1, j) \\ G_0(2, 1) \dots G_0(2, j) \\ G_0(3, 1) \dots G_0(3, j) \\ G_0(4, 1) \dots G_0(4, j) \\ G_0(5, 1) \dots G_0(5, j) \\ G_0(k, 1) \dots G_0(k, j) \end{array} \right] \dots (3)$$

5

In this case, in above Eq.(3), a suffix 0 of the gain data  $[G_{0 \times J}]$  denotes the flag data n (=0), x denotes the number of the channel, and J denotes the order 1...i...j of the frequencies set in the equalizers  $EQ_1$  to  $EQ_k$ .

10 In addition, in step S108, the gain data  $[G_{0 \times J}]$  are compared with predetermined threshold value  $THD_{CH}$  every channel, and sizes of the loudspeakers  $6_{FL}$  to  $6_{WF}$  on respective channels are decided based on the comparison results. That is, since the sound pressure of the reproduced sound reproduced by the  
 15 loudspeaker is changed according to the size of the loudspeaker, the sizes of the loudspeakers on respective channels are decided.

As the concrete deciding means, if the size of the loudspeaker  $6_{FL}$  on the first channel is decided, an average value  
 20 of the gain data  $G_0(1,1)$  to  $G_0(1,j)$  on the first channel in above Eq.(3) is compared with the threshold value  $THD_{CH}$ . If the average value is smaller than the threshold value  $THD_{CH}$ , the loudspeaker  $6_{FL}$  is decided as the small loudspeaker. Then, if the average value is larger than the threshold value  $THD_{CH}$ ,  
 25 the loudspeaker  $6_{FL}$  is decided as the large loudspeaker. In

in step S20 will be carried out. Such channel-to-channel level correcting process will be carried out in compliance with a flowchart shown in FIG.10.

First, the initialization process in step S200 is 5 executed, and the noise signal DN from the noise generator 3 can be input by switching the switch elements  $SW_{11}$  to  $SW_{51}$ . At this time, the switch elements  $SW_{11}$ ,  $SW_{k2}$  on the k-th channel are turned OFF. Also, the attenuation factors of the channel-to-channel attenuators  $ATG_1$  to  $ATG_k$  are set to 0 dB. 10 In addition, the delay times of all delay circuits  $DLY_1$  to  $DLY_5$  are set to 0. Further, the amplification factors of the amplifiers  $S_{1L}$  to  $S_{kF}$  shown in FIG.1 are made equal.

Besides, the frequency characteristic of the graphic equalizers  $GEQ$  is fixed to the state that they have been 15 adjusted by the above frequency characteristic correcting process.

Then, in step S202, the variable x representing the channel number is set to 1. Then, in step S204, the sound field characteristic measuring process is executed. The processes 20 in steps S204 to S208 are repeated until the sound field characteristic measurement of the channels 1 to 5 is completed.

Here, the noise signal (pink noise) is supplied in sequence to the equalizers  $EQ_1$  to  $EQ_k$  by exclusively turning ON the switch elements  $SW_{11}$ ,  $SW_{21}$ ,  $SW_{31}$ ,  $SW_{41}$ ,  $SW_{51}$  for the 25 predetermined period T respectively (steps S206, S208).

The microphone 8 collects respective reproduced sounds being reproduced by the loudspeakers  $6_{FL}$ ,  $6_{FR}$ ,  $6_C$ ,  $6_{RL}$ ,  $6_{RR}$  by this repeating process. Then, resultant sound collecting data  $DM$  ( $=DM_1$  to  $DM_5$ ) on respective channels are supplied to the 5 channel-to-channel level correcting portion 12. That is, the sound collecting data  $[DMX]$  represented by the matrix in following Eq. (6) are supplied to the channel-to-channel level correcting portion 12.

$$10 \quad [DBx] = \begin{bmatrix} DM1 \\ DM2 \\ DM3 \\ DM4 \\ DM5 \end{bmatrix} \quad \dots (6)$$

15 Then, after the measurement of the sound field characteristics on the first to fifth channels has been finished, the process goes to step S210. Then, one sound collecting data having the minimum value is extracted from the above sound collecting data  $DM_1$  to  $DM_5$ . Then, the extracted 20 result is set as the target data  $TG_{CH}$  for the channel-to-channel level correction.

Then, in step S212, adjusted values  $DM_1/TG_{CH}$ ,  $DM_2/TG_{CH}$ ,  $DM_3/TG_{CH}$ ,  $DM_4/TG_{CH}$ ,  $DM_5/TG_{CH}$ , used to adjust the attenuation factors of the channel-to-channel attenuators  $ATG_1$  to  $ATG_5$ , are 25 calculated by normalizing respective sound collecting data  $DM_1$ ,

to DM<sub>i</sub> in above Eq.(6) by the target data TG<sub>CH</sub>. Then, in step S214, the attenuation factors of the channel-to- channel attenuators ATG<sub>1</sub> to ATG<sub>5</sub> are adjusted by using the adjust signals SG<sub>1</sub> to SG<sub>5</sub> based on these adjusted values DM<sub>i</sub>/TG<sub>CH</sub> to DM<sub>5</sub>/TG<sub>CH</sub>.

5 With the above processes, the level adjustment between the first to fifth channels (x=1 to 5) except the k-th channel is completed.

10 In this fashion, the levels of the first to fifth channels are made proper by correcting the attenuation factors of the channel-to-channel attenuators ATG<sub>1</sub> to ATG<sub>k</sub> by virtue of the channel-to-channel level correcting portion 12.

15 Also, in the sound field characteristic measuring process in step S204, since resultant reproduced sounds are collected by sounding the loudspeakers 6<sub>FL</sub>, 6<sub>FR</sub>, 6<sub>C</sub>, 6<sub>RL</sub>, 6<sub>RR</sub> on time-division basis, the reproducing capabilities (output powers) of respective loudspeakers can be detected. Therefore, it is possible to achieve the total rationalization with taking account of the reproducing capabilities of respective loudspeakers.

20 Next, the phase characteristic correcting process in step S30 will be carried out in compliance with a flowchart shown in FIG.11.

25 First, the initialization process in step S300 is executed. The noise signal (pink noise) DN output from the noise generator 3 can be input by switching the switch elements

SW<sub>11</sub> to SW<sub>k2</sub>. Also, the frequency characteristics of the equalizers EQ<sub>1</sub> to EQ<sub>k</sub> are fixed to the already-adjusted characteristics as they are, and the attenuation factors of the channel-to-channel attenuators ATG<sub>1</sub> to ATG<sub>k</sub> are fixed as

5 they are, and also the delay times of the delay circuits DLY<sub>1</sub> to DLY<sub>k</sub> are set to 0. Furthermore, the amplification factors of the amplifiers S<sub>FL</sub> to S<sub>WF</sub> shown in FIG.1 are made equal.

Then, in step S302, the variable x representing the channel number is set to 1 and a variable AVG is set to 0. Then,

10 in step S304, the sound field characteristic measuring process is carried out to measure the delay times. Then, the processes in steps S304 to S308 are repeated until the sound field characteristic measurement of the first to k-th channels have been completed.

15 Here, the noise signal DN is supplied to the variable-gain filter portions BPF<sub>1</sub> to BPF<sub>5</sub> by exclusively turning ON the switch elements SW<sub>11</sub>, SW<sub>21</sub>, SW<sub>31</sub>, SW<sub>41</sub>, SW<sub>k1</sub> for the predetermined period T respectively.

According to this repeating process, the phase  
20 characteristic correcting portion 13 measures the noise sounds, that reach the listening position RV from the loudspeakers S<sub>FL</sub> to S<sub>WF</sub>, as the sound collecting data DM.

When this measurement has been completed, the process goes to step S310 wherein the phase characteristics of  
25 respective channels are calculated. Here, the cross

correlation between the sound collecting data DM measured when the noise signal DN is supplied to the first channel, i.e., a plurality of sound collecting data DM measured within the period T is calculated.

5        Then, a peak interval (phase difference) between resultant correlation values by the calculation is set as a delay time  $\tau_1$  of the first channel. Also, the delay times  $\tau_2$  to  $\tau_k$  are detected by calculating above similar cross correlations between the second to  $k$ -th channels.

10       Then, the process goes to step S312 wherein the variable AVG is incremented by 1. Then, in step S314, it is decided whether or not the variable AVG reaches a predetermined value AVERAGE. If NO, the processes starting from step S304 are repeated.

15       Here, the predetermined value AVERAGE is a constant indicating the number of times of the repeating processes in steps S304 to S312. In the present embodiment, the predetermined value AVERAGE is set to AVERAGE=4.

20       The delay times  $\tau_1$  to  $\tau_k$  of respective channels are calculated for every four channels by repeating the four times measuring process in this manner.

25       Then, in step S316, respective average values of every four delay times  $\tau_1$  to  $\tau_k$  are calculated. These average values  $\tau'_1$  to  $\tau'_k$  of respective delay times are set finally as the delay times.

Then, in step S318, the delay times of the delay circuits DLY<sub>1</sub> to DLY<sub>k</sub> are adjusted by the adjust signals SDL<sub>1</sub> to SDL<sub>k</sub> based on the finally calculated delay times  $\tau'_1$  to  $\tau'_k$ , whereby the phase characteristic correcting process has been completed.

5 In this manner, in the phase characteristic correcting process, the loudspeakers are sounded by supplying the pink noise from the graphic equalizer GEQ side, and then the delay times are calculated from the sound collecting results of resultant reproduced sounds. Therefore, the delay times of  
10 the delay circuits DLY<sub>1</sub> to DLY<sub>k</sub> are not simply adjusted (corrected) based on only the propagation delay times of the reproduced sounds, but it is possible to implement the total rationalization while taking account of the reproducing capabilities of respective loudspeakers and the  
15 characteristic of the audio system.

Next, the process in step S40 will be carried out in compliance with a flowchart shown in FIG.12.

First, in step S400, the noise signal (uncorrelated noise) DN output from the noise generator 3 can be input by  
20 switching the switch elements SW<sub>11</sub> to SW<sub>k1</sub>. Also, the frequency characteristics of the variable gain filter portion BPF<sub>1</sub> to BPF<sub>k</sub> are fixed to the already-adjusted characteristics, and the attenuation factors of the channel-to-channel attenuators ATG<sub>1</sub> to ATG<sub>k</sub> are also fixed as they are. Then, the delay times  
25 of the delay circuits DLY<sub>1</sub> to DLY<sub>k</sub> are fixed to the

already-adjusted delay times. Further, the amplification factors of the amplifiers  $5_{FL}$  to  $5_{WF}$  shown in FIG.1 are made equal.

Then, in step S402, the attenuation factor of the channel-to-channel attenuator  $ATG_k$  on the k-th channel is set 5 to 0 dB.

Then, in step S404, the noise signal (uncorrelated noise) DN is simultaneously supplied to the first to fifth channels except the k-th channel.

Accordingly, the all frequency band loudspeakers  $6_{FL}$ ,  $6_{FR}$ , 10  $6_C$ ,  $6_{RL}$ ,  $6_{RR}$  are simultaneously sounded by the noise signal DN in the all frequency band, and then the flatness correcting portion 14 receives resultant sound collecting data DM.

In step S406, the flatness correcting portion 14 calculates the power spectrum  $P_L$  of the reproduced sound 15 reproduced in the low frequency band by the all frequency band loudspeakers  $6_{FL}$  to  $6_{RR}$  and the power spectrum  $P_M$  of the reproduced sound in the middle/high frequency band by spectrum-analyzing the sound collecting data DM.

Then, in step S408, the noise signal (pink noise) DN is 20 supplied only to the k-th channel.

Accordingly, only the low frequency band exclusively reproducing loudspeaker  $6_{WF}$  is sounded by the noise signal DN in the low frequency band, then the flatness correcting portion 14 receives resultant sound collecting data DM in the low 25 frequency band.

Then, in step S410, the flatness correcting portion 14 calculates the reproduced sound power  $P_{WFL}$  reproduced by the low frequency band exclusively reproducing loudspeaker  $6_{WF}$  in the low frequency band by spectrum-analyzing the sound 5 collecting data DM in the low frequency band.

Then, in step S412, the flatness correcting portion 14 generates the adjust signal  $SG_k$  by executing the calculation expressed by following Eq.(7) to adjust the attenuation factor of the channel-to-channel attenuator  $ATG_k$ .

10 
$$SG_k = (TG_L \times P_{MH} - TG_{MH} \times P_L) / TG_{MH} \times P_{WFL} \quad \dots (7)$$

A coefficient  $TG_{MH}$  in above Eq.(7) is an average value of the target curve data corresponding to the middle/high frequency band, out of the target curve data which the listener selects among the target curve data [TGx] shown in above Eq.(1) 15 or the default target curve data which the listener does not select. Also, a coefficient  $TG_L$  is an average value of the target curve data corresponding to the low frequency band.

Then, in step S414, the attenuation factor of the channel-to-channel attenuator  $ATG_k$  is adjusted by using the 20 adjust signal  $SG_k$ , whereby the automatic sound field correcting process has been completed.

In this manner, in the case that the audio sound is reproduced by all frequency band loudspeakers  $6_{FL}$ ,  $6_{FR}$ ,  $6_c$ ,  $6_{RL}$ ,  $6_{RR}$ ,  $6_{WF}$ , the frequency characteristic of the reproduced sound 25 in the sound field space can be made flat over the full audio

frequency range if the level correction is executed finally between the channels by the flatness correcting portion 13. Therefore, the problem in the prior art such as the increase of the low frequency band level shown in FIG. 6 can be overcome.

5 Also, in the sound field characteristic measuring process in steps S404 to S408, since the reproduced sounds generated by sounding respective loudspeakers  $6_{FL}$ ,  $6_{FR}$ ,  $6_C$ ,  $6_{RL}$ ,  $6_{RR}$ ,  $6_{WF}$  on time-division basis are collected, the reproducing capabilities (output power) of respective loudspeakers can be 10 detected. Therefore, the total rationalization with taking the reproducing capabilities of respective loudspeakers into consideration can be achieved.

Then, the audio signals  $S_{FL}$ ,  $S_{FR}$ ,  $S_C$ ,  $S_{RL}$ ,  $S_{RR}$ ,  $S_{MF}$  from the sound source 1 are set into the normal input state by turning 15 OFF the switch element SW<sub>N</sub>, turning OFF the switch elements SW<sub>11</sub>, SW<sub>21</sub>, SW<sub>31</sub>, SW<sub>41</sub>, SW<sub>51</sub>, SW<sub>k1</sub> connected to this switch element, and turning ON the switch elements SW<sub>12</sub>, SW<sub>22</sub>, SW<sub>32</sub>, SW<sub>42</sub>, SW<sub>52</sub>, SW<sub>k2</sub>, and thus the present audio system is brought into the normal audio playback state.

20 As described above, according to the present embodiment, since the characteristics of the sound field space at the listening position RV are corrected while totally taking account of the characteristics of the audio system and the loudspeakers, the extremely high quality sound field space with  
25 the presence can be provided.

Also, the correction to implement the very high quality sound field space with the presence is made possible by executing the sound field correcting process in the order of steps S10 to S40 shown in FIG.8.

5 In the present embodiment, the automatic sound field correcting system of the so-called 5.1 channel multi-channel audio system that includes the wide frequency range loudspeakers 6<sub>PL</sub> to 6<sub>RR</sub> for five channels and the low frequency band exclusively reproducing loudspeaker 6<sub>WF</sub> has been explained,  
10 10 but the present invention is not limited to this. The automatic sound field correcting system of the present invention can be applied to the multi-channel audio system that includes the loudspeakers that are larger in number than the present embodiment. Also, the automatic sound field  
15 15 correcting system of the present invention can be applied to the audio system that includes the loudspeakers that are smaller in number than the present embodiment.

The sound field correction in the audio system including the low frequency band exclusively reproducing loudspeaker 20 (subwoofer) 6<sub>WF</sub> has been explained, but the present invention is not limited to this. The high quality sound field space with the presence can be provided by the audio system including only the all frequency band loudspeakers without the subwoofer. In this case, all channel characteristics may be corrected by  
25 25 the channel-to- channel level correcting portion 12 not to use

the flatness correcting portion 14.

In the present embodiment, in step S412 shown in FIG.12, as apparent from above Eq.(7), the rationalization of the attenuation factor of the channel-to-channel attenuator ATG<sub>K</sub> 5 is performed on the basis of the levels of the reproduced sounds of all frequency band loudspeakers 6<sub>FL</sub> to 6<sub>RR</sub>. That is, the levels of the reproduced sounds of all frequency band loudspeakers 6<sub>FL</sub> to 6<sub>RR</sub> are used as the basis by setting a product 10 of the target data TG<sub>MH</sub> in the middle/high frequency band and the variable P<sub>WF</sub>, that corresponds to the level of the reproduced sound of the low frequency band exclusively 15 reproducing loudspeaker 6<sub>WF</sub>, in the denominator of above Eq.(7). However, the present invention is not limited to this. The rationalization of the attenuation factors of the channel-to-channel attenuators ATG<sub>1</sub> to ATG<sub>K</sub> is performed on the basis of the level of the reproduced sound of the low frequency band exclusively reproducing loudspeaker 6<sub>WF</sub>.

That is, in the present embodiment, the flatness correcting portion 14 corrects the attenuation factor of the channel-to-channel attenuator ATG<sub>K</sub>. Conversely, the level of the reproduced sound of the low frequency band exclusively reproducing loudspeaker 6<sub>WF</sub> may be measured, then the attenuation factor of the channel-to-channel attenuator ATG<sub>K</sub> 20 may be set on the basis of measured result, and then the attenuation factors of the channel-to-channel attenuators ATG<sub>1</sub> 25

to ATG<sub>1</sub> may be corrected on the basis of the attenuation factor of the channel-to-channel attenuator ATG<sub>k</sub>.

Further, as shown in FIG.2, each of the signal transmission lines of respective channels are constructed by

5 connecting the band-pass filters, the inter-band attenuators, the adder, the channel-to-channel attenuator, and the delay circuit in sequence following to the graphic equalizer GEQ. However, such configuration is shown as the typical example and thus the present invention is not limited to such 10 configuration.

For example, the channel-to-channel attenuator ATG<sub>1</sub> to ATG<sub>k</sub> and the delay circuit DLY<sub>1</sub> to DLY<sub>k</sub> may be arranged prior to the graphic equalizer GEQ, otherwise the graphic equalizer GEQ may be arranged between the channel-to- channel attenuator

15 ATG<sub>1</sub> to ATG<sub>k</sub> and the delay circuit DLY<sub>1</sub> to DLY<sub>k</sub>.

The reasons for enabling the configuration of the present invention to change appropriately the positions of the constituent elements are that, unlike the conventional audio

system in which the correction of the frequency characteristic and the correction of the phase characteristic are performed

20 respectively by separating respective constituent elements, the noise signal from the noise generator can be input from the input stage of the sound field correcting system and also the frequency characteristic and the phase characteristic of 25 the overall sound field correcting system can be corrected

totally. As a result, the automatic sound field correcting system of the present invention makes it possible to correct properly the frequency characteristic and the phase characteristic of the overall audio system and to enhance 5 margin in design.

Also, upon correcting the attenuation factors of the channel-to-channel attenuators by the flatness correcting portion 14, the pink noise is supplied from the noise generator 3 to the loudspeaker 6<sub>WF</sub>. But other noises may be supplied.

10 As described above, according to the automatic sound field correcting system according to the present invention, since the sound field correction is performed while taking totally account of the characteristics of the audio system and the loudspeakers, the extremely high quality sound field space 15 with the presence can be provided.

Also, in the audio system including the low frequency band exclusively reproducing loudspeaker and wide frequency band loudspeakers, since a new function for making the level of the low frequency band reproduced sound and the level of 20 the middle/high frequency band reproduced sound equal is provided, the extremely high quality sound field space with the presence can be provided.